NOISE ACTIVATED AUTOMATIC VOLUME CONTROL

by

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THESIS

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ABSTRACT

Communications operators who must use receivers in a noisy audio environment without earphones are plagued by the effect of a varying background noise level. Often the operators are simultaneously performing other tasks thus can only monitor voice circuits. As the noise level varies the operator must repeatedly adjust the volume control for a comfortable listening level.

An analysis of the physical situation is presented. An automatic feedback control system that measures the predominate range of frequencies of the background noise and automatically adjust the volume control of a power amplifier for a satisfactory listening level is developed. This system relieves the operator of the task of manual volume control and reduces the effect of noise masking.

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LIST OF SYMBOLS

A	Gain of the audio power amplifier. A function of frequency.
B	Reference Bias used in the summing amplifier to set the operating point of the AGC unit
C	Control voltage used for AGC to set the gain of the control unit. A function of B plus the difference between the determined average signal and average signal-plus-noise levels.
H	Transfer function for the control unit and cascade audio power amplifier. A function of frequency and the control voltage.
L	Transfer function of the signal-level sampling channel. A function of frequency.
M	Transfer function of the microphone channel. A function of frequency.
N	Noise. Any audio frequency vibration present in the area that is not the desired intelligence signal.
P	Level of the signal S after amplification, as presented to the loudspeaker.
R	Transfer function representing the combined characteristics of the loudspeaker and acoustical properties of the room in which the system operates.
S	Input level of the desired intelligence signal to the control unit.

I. INTRODUCTION

In many aspects of life today the human operator performs multiple tasks under varying conditions. On the bridge of a ship a command-communications receiver may be monitored by the OOD while he performs other tasks. In a small boat the coxswain may also monitor a command voice circuit. In a classroom or lecture hall the student listens to the lecturer and takes notes.

Each of these cases shares a common problem, that of environmental noise. For the coxswain in the small boat, his noise source is the main engines. For the student, a passing aircraft can drown out the speaker. The coxswain can manually adjust the volume level on his speaker if he is not engaged in maneuvering the craft. The lecturer usually cannot change the level of the public-address system. The OOD on the bridge can readjust his speakers.

In all cases repeated action by the operator is necessary to provide a comfortable listening level in the presence of a varying background noise. If the operator does not react quickly enough to large scale fluctuations in the background level he may either miss message parts or develop fatigue from listening to a loudspeaker which operates at a higher than necessary volume level.

An automatic control system that senses the background noise level and adjust the speaker level can prevent loss of messages and lessen fatigue. This paper presents a feedback control system that

monitors the sound level and electronically adjusts the gain of a control unit so that the signal is at the minimum comfortable level necessary for the intelligence to be understood over the background noise level.

In the following discussions, the term signal refers to the desired intelligence signal; noise refers to any audio sounds that are not part of the intelligence signal. Signal plus noise means a combination of both the desired intelligence and background noise.

II. ANALYSIS OF THE PHYSICAL SITUATION

An automatic control system that monitors noise level in a space and adjusts a signal source in that space can easily be unstable. If the control system detects the signal source and interprets it as noise and increases the signal level, the apparent noise level (signal-plus-background noise) increases. The signal level would then increase continuously until the system operated at its maximum volume. Some mechanism is required that will cancel the signal level from the total signal-plus-noise level if an appropriate measure of the background noise level is to be made.

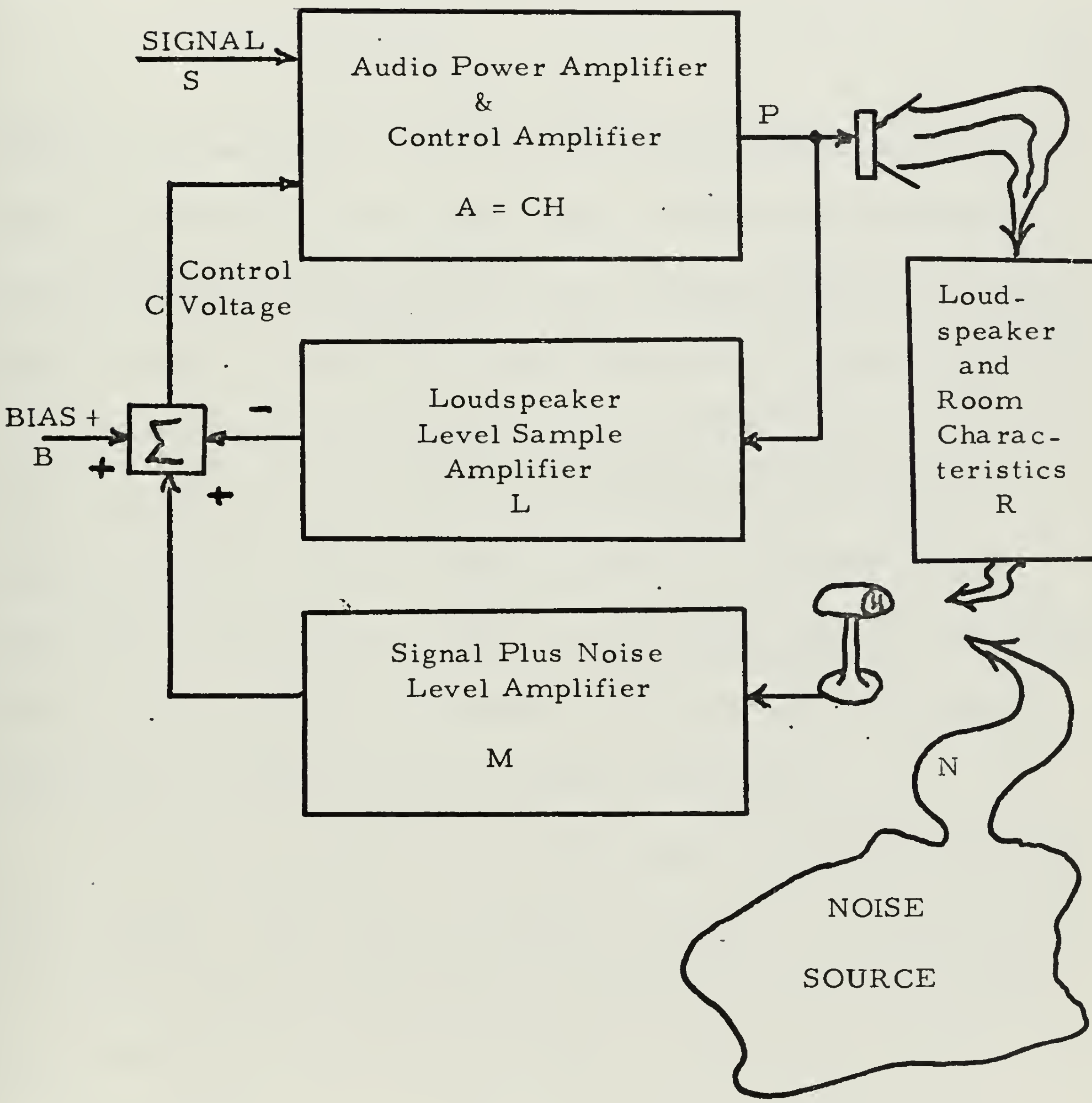
Figure 1. indicates a possible block diagram for the automatic control system. It is desirable that the audio level of the signal be a function only of some reference bias B and background noise level N. The desired signal P should not affect the audio output level. The signal level is a function of the power amplifier gain, A; the input signal level, S; and the acoustic propagation to the listener. The amplifier gain A can be a linear function of the control voltage, C. In this case

$$A = C \cdot H \tag{1}$$

where H is the power amplifier transfer function with gain proportional to C. The level of the audio signal fed to the loudspeaker can then be expressed as

$$P = C \cdot H \cdot S \tag{2}$$

Figure 1
Block Diagram of the Feedback Control
System Transfer Functions



Considering the loudspeaker efficiency to be a part of the propagation losses in the room characteristics transfer function, R, the control voltage can be expressed in terms of the other parameters from figure 1:

$$C = B + N \cdot M + P \cdot (R \cdot M - L) \tag{3}$$

Substituting equation (2) and rearranging equation (3)

$$C = \frac{B + N \cdot M}{1 - S \cdot H \cdot (R \cdot M - L)} \tag{4}$$

Where L represents the transfer function of the amplifier whose output is proportional to the average signal level to the loudspeaker. M is the transfer function whose output is proportional to the sensed noise level as processed by the microphone and microphone amplifier.

According to both equations (3) and (4), when L equals R· M the control parameter C is independent of the input signal and of the output amplified signal P. Likewise when L is greater than R· M under compensation occurs; while, when L is less than R· M the signal influences the control in an adverse way causing the system to increase the level until the power amplifier saturates.

The transfer functions M, L, and A can be selected in the design of the system. The transfer function R, however, is dependent on the acoustical properties of the space in which the system is installed, and on the alignment of the loudspeaker and microphone sensor.

It is intuitively obvious that if the speaker radiated into an unenclosed space no reflections would return to the microphone sensor.

If a directional microphone were used and oriented so that it could not pick up direct speaker radiation, then R would be effectively zero. Such an installation may be closely approximated in an open cockpit boat and on an outdoor lecture platform. In a more practical case the speaker is enclosed in a small room with hard acoustical properties (long reverberation time).

In a small acoustically hard room, radiation from the loudspeaker travels by multiple paths to the microphone sensor. The microphone, being a pressure-to-electrical-signal transducer, is sensitive to the phase characteristics of the pressure wave it experiences. Multipath propagation of the acoustical wave can produce standing-wave patterns for each of the frequency components in the room. The standing wave pattern is a function of the acoustical properties of the room, and of the loudspeaker and microphone radiation pattern. These factors influence the gain-frequency characteristics of the transfer function R.

If the propagation medium for the acoustic wave (air) maintained a constant velocity of propagation then R could be analytically described. However with the inclusion of an operator or other persons who are free to move about in the room, a random variation factor is introduced. The position of the operator affects the standing-wave patterns. Variations in the density of air change the velocity of propagation and alters the standing-wave patterns. Under these

conditions the complex nature of R cannot be specified in a simple form but must be statistically described.

Fortunately the air density variations are small over a short time period. The random variation effects caused by the operator or other moving persons can be reduced by aligning the loudspeaker so that it fills the room with the signal without directly radiating the operator. This will minimize the operator's effect on the standing-wave pattern. With these constraints R becomes predominately a function of frequency only. The magnitude of the product $R \cdot M$ averaged over the range of signal frequencies present can then be made nearly equal to the magnitude of L.

III. SELECTION OF THE TRANSFER FUNCTIONS

The transfer functions L, M, and A are controllable in the design; but, R is not completely controllable. Since A processes the signal with gain variation only, it should have a flat response across the band of expected signals.

For most military receivers the necessary audio bandwidth extends from 300 Hertz to 3300 Hertz. In a public address system better fidelity is available with somewhat more bandwidth. Standard AM commercial broadcasts contain audio frequencies from 100 to 5000 Hertz while commercial FM broadcasts go from 50 to 15,000 Hertz.

For a general case the control unit transfer function was selected to extend from 45 to 20,000 Hertz. With this wideband response the control unit could be used for any audio application from military communications to hi-fi entertainment systems. The actual form of A would then be determined completely by the power amplifier in the receiver.

The frequency response of the microphone channel, M, is primarily controlled by the response of the microphone. The microphone used has the frequency response shown in figure 2. Better quality microphones have responses that are less peaked and more broadband.

By designing the frequency response of the microphone amplifier to have 3dB cutoff frequencies below 20 and above 20,000 Hertz, M

is completely described by the microphone's frequency response. However this is not necessarily the most desirable attack, particularly if the microphone has a peaked response coincident with a predominate noise frequency.

A better approach considers the effect of noise masking. The presence of noise containing low frequency components raises the minimum discernible level for audio signals of the same frequency and higher in frequency. High frequency noise has little masking effect on lower frequency audio signals.[1]

By design of the transfer function M, the microphone response and microphone-amplifier-channel response can apply an appropriate weighting factor to the sampled noise frequencies. An example of an arbitrary wideband weighting function is shown in figure 3. An optimum weighting function requires exact knowledge of the noise spectrum expected and the frequency range of the signal.

The modification of R is somewhat limited by the nature of the communications installation. The primary consideration is to remove the large scale variation in the signal fed back to the microphone. This is accomplished by the placement of the speaker at a higher level than the operator and directed toward the ceiling. Softening the acoustical properties of the space by the use of drapes or sound absorbing panels is desirable but not practical on naval vessels.

Since M is modified by R the exact specification of M is meaningless without an appropriate representation of R. Thus in the system

Figure 2
Frequency Response of Shure
Model 570 Dynamic Microphone

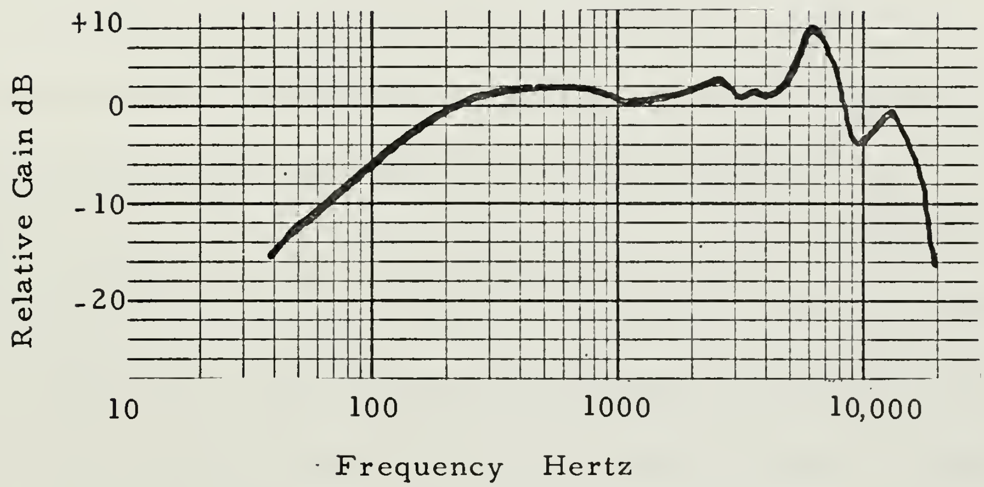
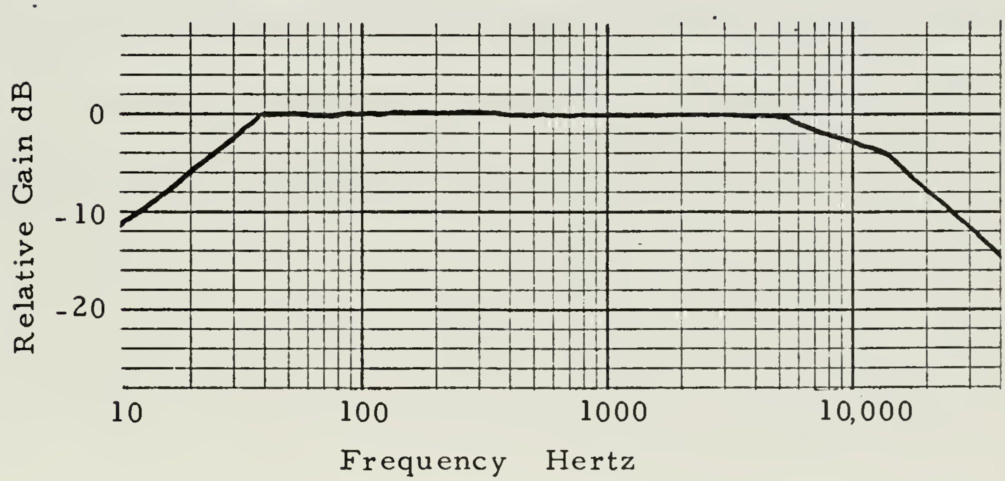


Figure 3
Wideband Weighting Function for Microphone
Channel Considering Human Ear Response
and Sound Masking



designed for testing the response used provided maximum weighting for all frequencies between 200 and 6500 Hertz. The low end was controlled by the microphone response. The high end provided some attenuation of the microphone's peaked response at 6500 Hertz, and greater attenuation of frequencies above 12 kilohertz. This wide-band response proved suitable for the general case where the interfering noise frequencies were anywhere in the audio spectrum.

Since L must equal R$\cdot$M if the noise level is to have exclusive control of the system, L should also have the same transfer function as R$\cdot$M. Practically, since no new frequencies are generated by R, L can have any high frequency cutoff greater than or equal to A; and, a low frequency cutoff lower than or equal to A.

M can have a high-frequency roll off at a lower frequency than L to provide weighting without loss of system stability. In a voice program the majority of the sound energy is concentrated in the lower frequencies. In a musical concert more high frequencies are present; but, these are lost in propagation more quickly than the lower frequencies. Thus the average energy level can be determined by the averaging of the low frequencies.

IV. THE CONTROL SYSTEM

A. BASIC CONSIDERATIONS

To verify the concepts discussed in previous sections, a control system was designed and built with the following objectives:

1. It must be compatible with existing sound equipment.
2. It must be easily installed.
3. It must not disturb the proper operation of the equipment in which it is installed.
4. It must make use of the existing power amplifier.
5. It must be relatively easy to align.
6. It must be stable over a wide range of operating temperatures.
7. It must have low power drain.

Condition 1 required consideration of the type of equipment in which this unit could be installed. By designing the control system for operation from 12 volt power supplies, the unit can be used in all types of communications receivers including portable receivers that operate off automobile batteries.

Condition 2 required consideration of incorporation in existing equipment. Automatic-gain-control circuits provide the greatest dynamic range when used with low level signals. In most receivers AGC is provided to vary the gain of the IF stages according to the strength of the incoming RF signal. Most receivers also employ a fixed-gain audio amplifier. Audio level is controlled by a volume

control potentiometer that adjusts the size of the input signal to the audio amplifier. This control usually provides the load on the detector stage. By placing the control unit between the tap of the manual volume control and the input to the audio amplifier, the control unit can act as an electronic potentiometer. Instead of manually increasing the size of the signal applied to the audio amplifier, the control system computes a bias for the control stage. This bias adjusts the stage gain. Thus a larger signal is fed to the audio amplifier when high ambient noise is present. This method of adjusting the signal level has the advantage of not interferring with the AGC applied to the IF stages of the receiver. Low level signals are also applied to the control unit for greatest dynamic range. Moreover, some degree of volume control is still available through the use of the manual control. Additional control is afforded by the bias control in the summing amplifier. This insertion location of the control amplifier satisfied conditions 3 and 4.

Alignment consists of balancing the gains of the sample and microphone channels for the full range of expected noise levels. With proper balance the output level does not change due to variations in the signal level.

Condition 6 required consideration of the operating environment. Wide temperature ranges are encountered even on board boats that operate in a temperate climate such as in New York Harbor. Summer temperatures can reach 100 degrees Fahrenheit. Winter

temperatures dip to freezing and below. This suggests the use of integrated circuits that are temperature stabilized from -25 to 125 degrees Centigrade. Integrated circuits also satisfy the condition for low power drain. A block diagram of the system as constructed is shown in figure 4.

In the test system the total current drain was about 70 mA. Since integrated circuits were used, a compromise was made whereby both positive and negative 12 volt supplies were necessary. Each power source supplied about 35 mA. Minor redesign of the test system could reduce the current requirement and the requirement for both positive and negative voltage sources.

B. THE GAIN CONTROL STAGE

A two-stage RC-coupled integrated-circuit DC amplifier (the CA3000) was used in the gain-control stage. Figure 5 shows the basic circuit diagram. The CA3000 is a differential amplifier employing emitter follower inputs and constant-current sink in the common-emitter lead of the differential pair. By controlling the bias voltage on the current-sink transistor, a wide range of voltage gain is obtainable. Each stage has a maximum voltage gain of 32dB. By cascading two stages with a small amount of negative feedback to limit the maximum gain, overall gain was controllable from -6 to 55 dB. The bias voltage necessary to provide this gain variation is plotted in figure 6. A linear range of 34 dB is available using AGC

Figure 4
Block Diagram of the Automatic Control System

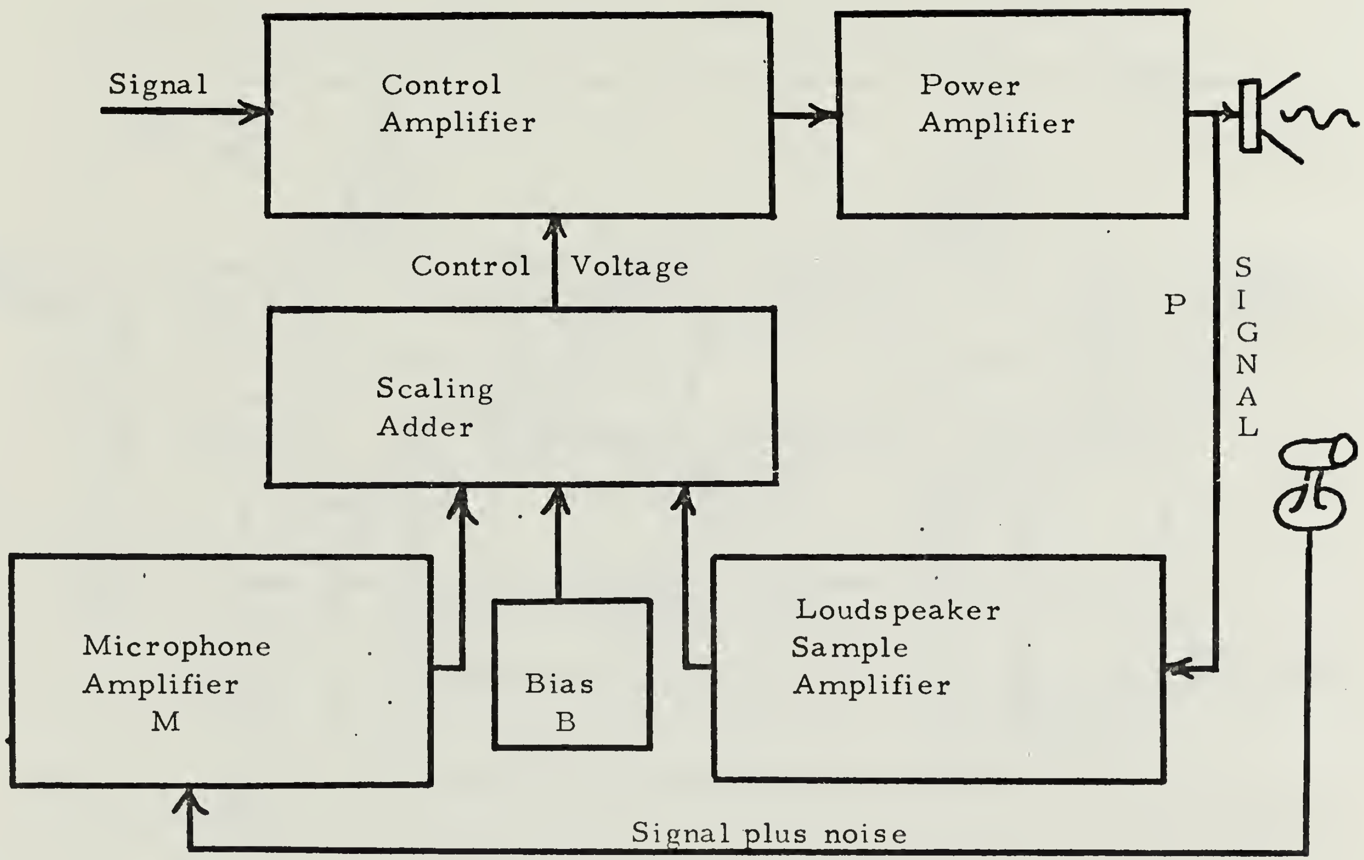


Figure 5
Schematic Diagram of the Gain Control Unit

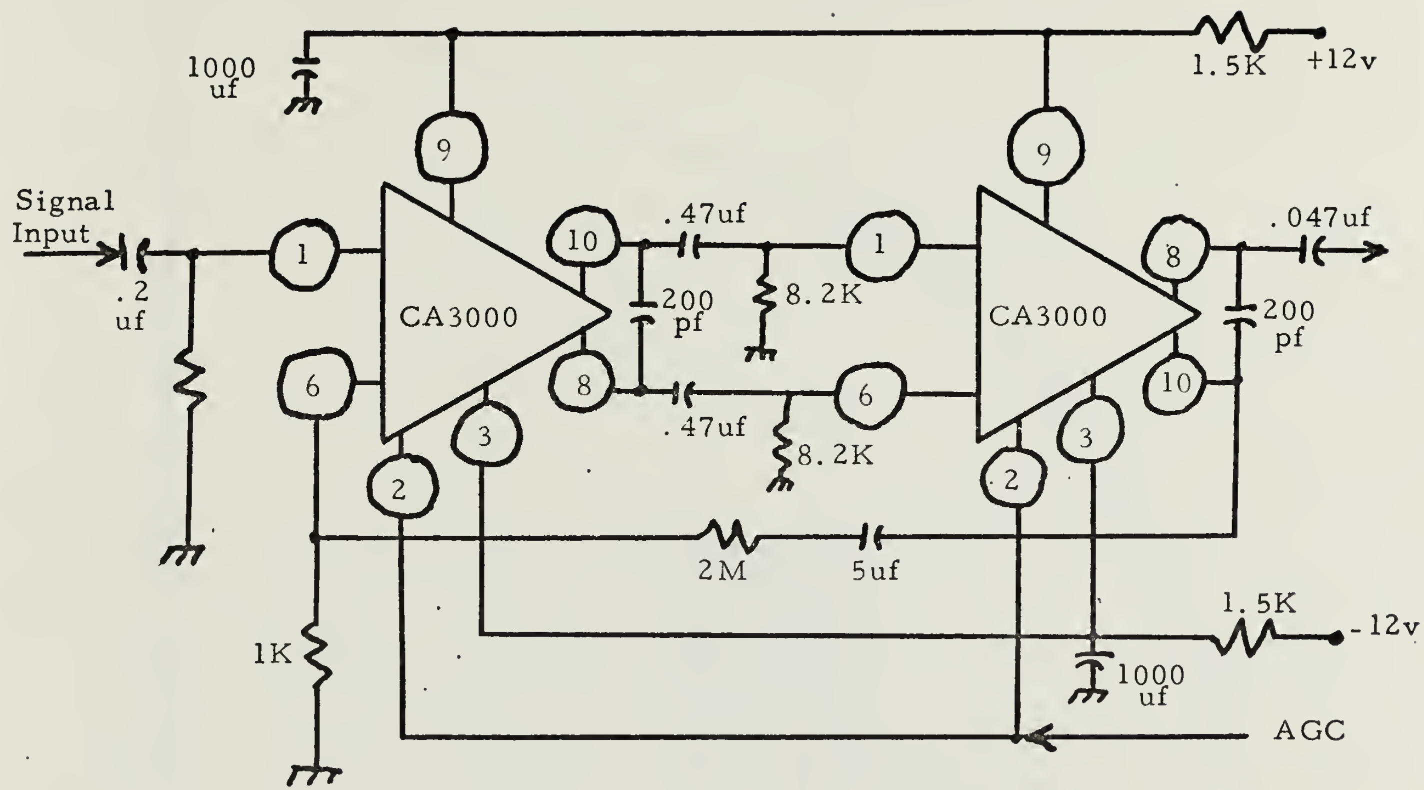
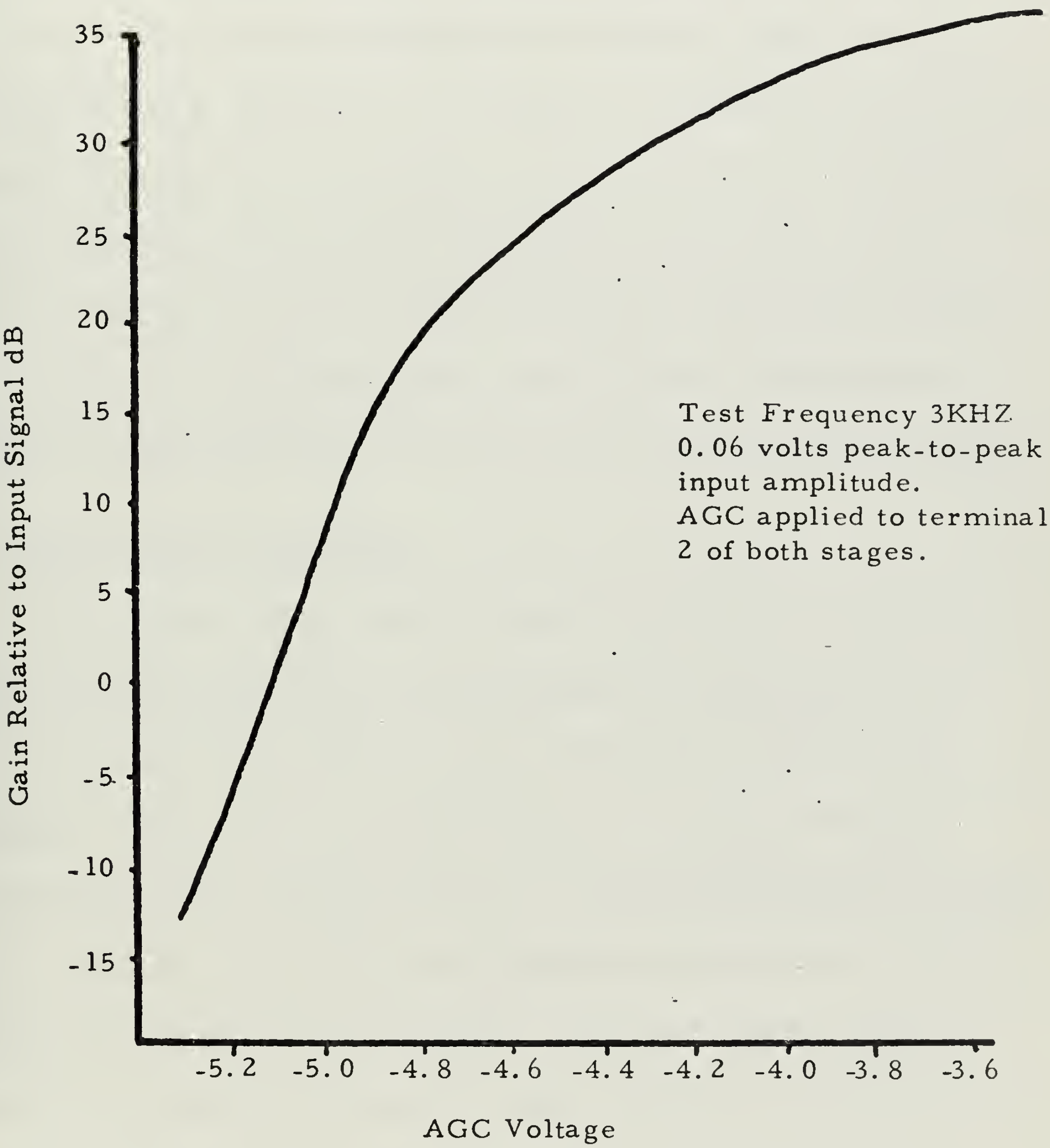


Figure 6
Voltage Gain Across the 2 Stage
Control Amplifier



on both stages. The input stage required a maximum input signal no larger than 0.1 volt peak-to-peak to operate over its entire range with no distortion. This is a level easily obtainable from the volume-control potentiometer in a radio receiver. The input terminal is at ground potential since both positive and negative balanced supplies are used. This can be an advantage if the volume control potentiometer is also at DC ground. The frequency response of the control stage is controlled by the coupling capacitors and the 200 picofarad shunt capacitor across the push-pull outputs. The values selected give a frequency response as shown in figure 7. The 3 dB bandwidth is from 45 to 22,000 Hertz, wide enough to handle anything from military communications receivers to hi-fi entertainment applications.

C. THE SAMPLE CHANNEL

The network that samples the audio output level picks off the signal from the feed wire to the loudspeaker. At this point any fluctuation in power amplifier output is sensed as it goes to the speaker. This circuit is shown in figure 8. By going through a potentiometer this high-level signal is reduced then amplified through two stages. By using a two-stage amplifier the magnitude of the gain function L is more easily controlled. Also the amplifier can be adjusted to provide large signals without entering its saturation region. This maximum voltage swing can be set according to the largest output level of the power amplifier.

Figure 7
Frequency Response of the Control Unit

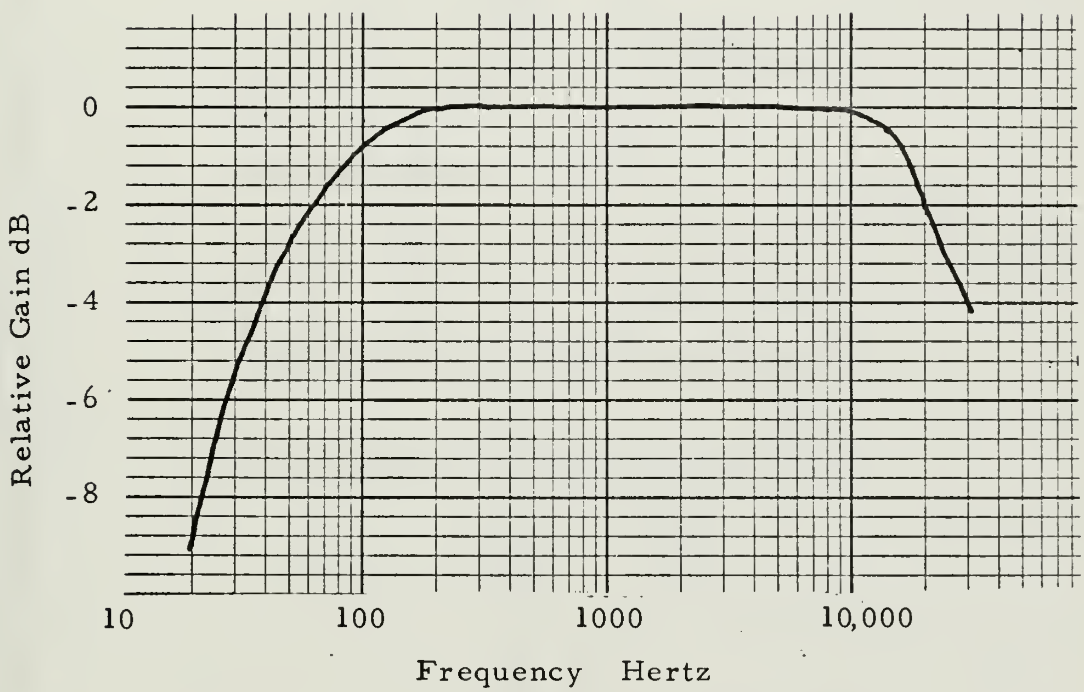
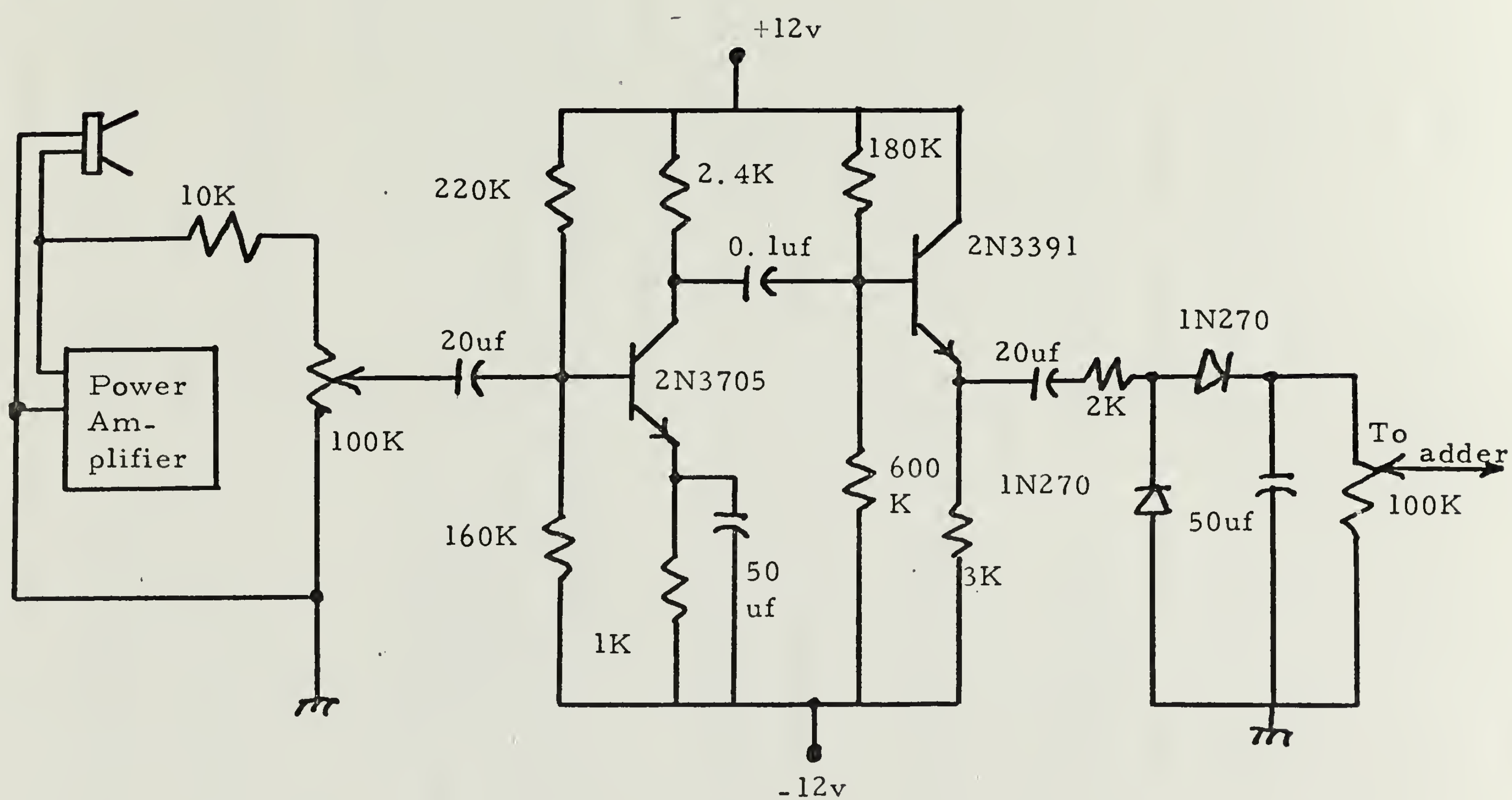


Figure 8
Schematic Diagram of Speaker Level Sample Channel
and Peak Detector



Bias in both stages was carefully selected to give maximum peak-to-peak voltage swing capability. The emitter-follower feeds a peak detector which clamps the waveform positive and charges the 50 microfarad capacitor with a positive voltage approximately proportional to the average value of the sampled waveform. The emitter follower is used because of its low output impedance. Thus, the attack time constant is controlled by the 2 kilohm resistor that feeds the peak detector. An attack time constant of 0.1 second was chosen to provide both quick response to sudden changes in signal level and a measure of impulse noise suppression.

The 100 kilohm potentiometer bleeds the 50 microfarad capacitor to provide a release time constant of 5 seconds which is suitable for both music and voice programs.

The frequency response of the sample amplifier from the input to the detector is shown in figure 9. The wide-band response matches the response of the control unit, thus is suitable for any type of program material.

D. THE MICROPHONE CHANNEL

One of the more critical aspects of the control system is the microphone channel shown in figure 10. This channel senses the total signal-plus-noise level present in the space. After sufficient amplification, peak detection is applied and the signal averaged. The detection and averaging circuit is the same as that used in the

Figure 9
Frequency Response of the Speaker Level
Sampling Channel

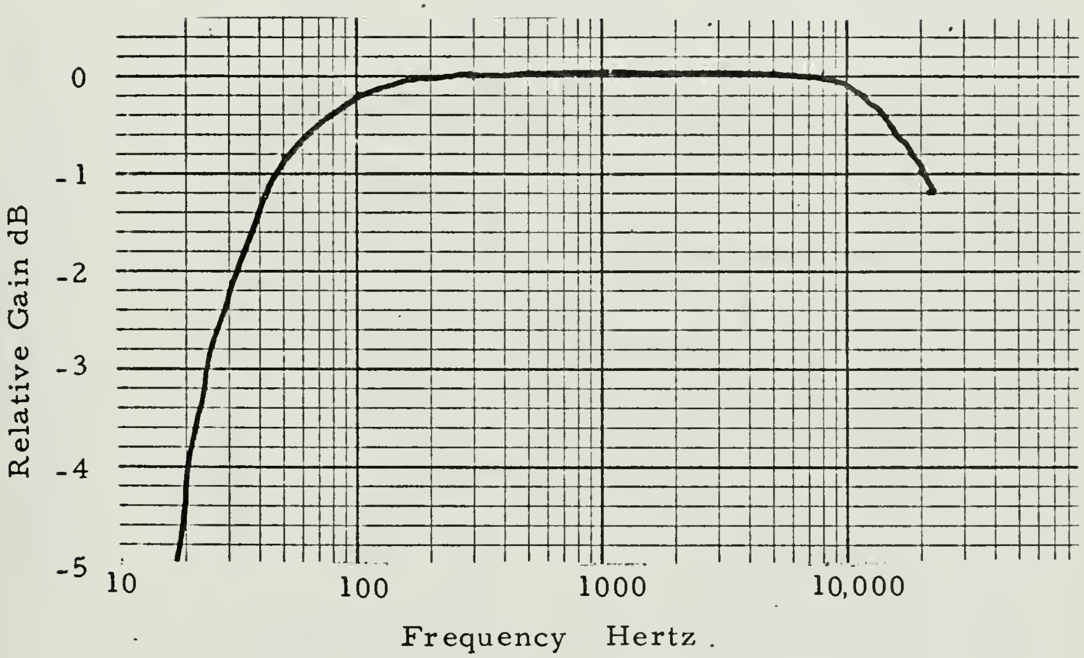
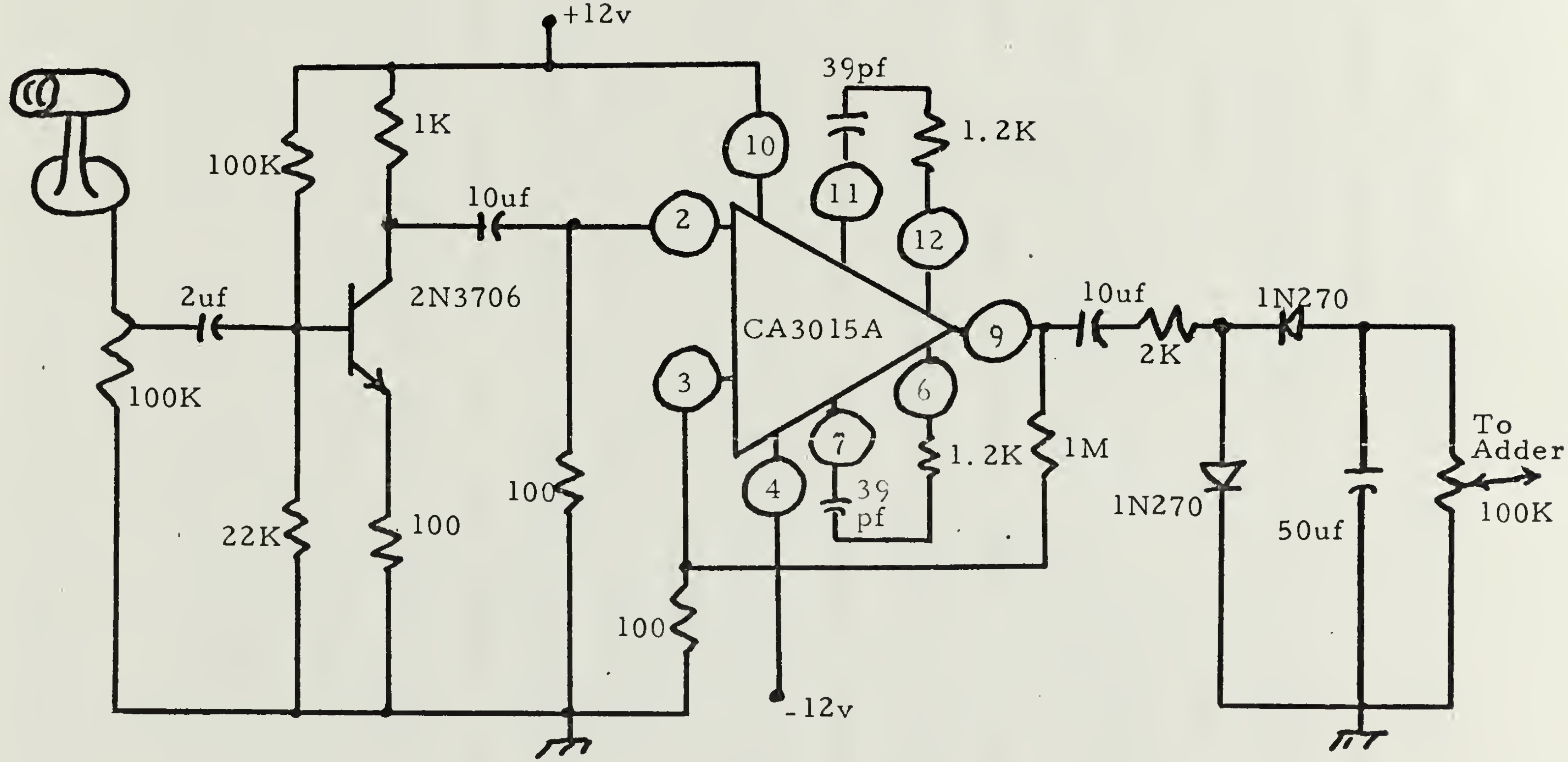


Figure 10
Schematic Diagram of the Microphone
Amplifier and Peak Detector



sample circuit with the exception of negative clamping and a slightly longer release time constant.

The positive and negative averaged voltages are added in the operational-amplifier scaling adder. The longer time constant provides a better system transient response. At the termination of a period of loud background noise level, the gain of the control stage decays according to the release time constants. By having the noise release time slightly longer than the signal release time the audio output is maintained at a higher level during decay. This enables the decay to match the recovery of the ear's sensitivity to lower levels.

The first transistor in the microphone channel provides impedance matching between the microphone and the low input impedance of the operational-amplifier second stage. Some voltage gain is also provided. The noise generated in this stage is amplified by the high-gain second stage and aids in the noise detection process as follows. The peak detector requires input signals larger than 0.3 volt peak-to-peak to bias the 1N270 diodes in the on state. Without this bias the peak detector would not provide any output until its input signal reached the 0.3 volt. The control system would provide no control for low-level noise signals. However the diodes can be biased on either with an external biasing arrangement or by the noise generated in the first stage. The stage noise in this case was sufficient to bias the diodes in the on state.

The second stage CA3015A operational amplifier is used in a narrow-band high-gain mode. It provides an additional 70 dB of gain. With this operational amplifier the maximum voltage swing available is 14 volts peak-to-peak. Large signals are used so that the germanium diode turn-on thresholds of 0.3 volt will not reduce the linearity at low levels.

The frequency response of the microphone channel from the input to the peak detector is shown in figure 11. By providing nearly flat response down to 60 Hertz (3 dB cutoff below 20 Hertz) the low end of the response for M is controlled entirely by the characteristics of the microphone. The upper frequency level is rolled off early to provide weighting to the lower frequencies where masking has the most influence. High-frequency noise above 12 kilohertz is not critical due to the reduced response of the ear at these higher audio frequencies.

E. THE SCALING ADDER

An operational amplifier (CA3015A) is used as an inverting scaling adder. The connections are shown in figure 12. The inputs are tapped off the outputs of the microphone channel, the sample channel, and a bias reference. The 1 megohm input resistors and 100 kilohm feedback resistors provide for a divide-by-ten linear scale factor. A variable scaling factor and isolation is also introduced by the potentiometers. The output is a negative bias driven

Figure 11
Frequency Response of the
Microphone Amplifier Channel

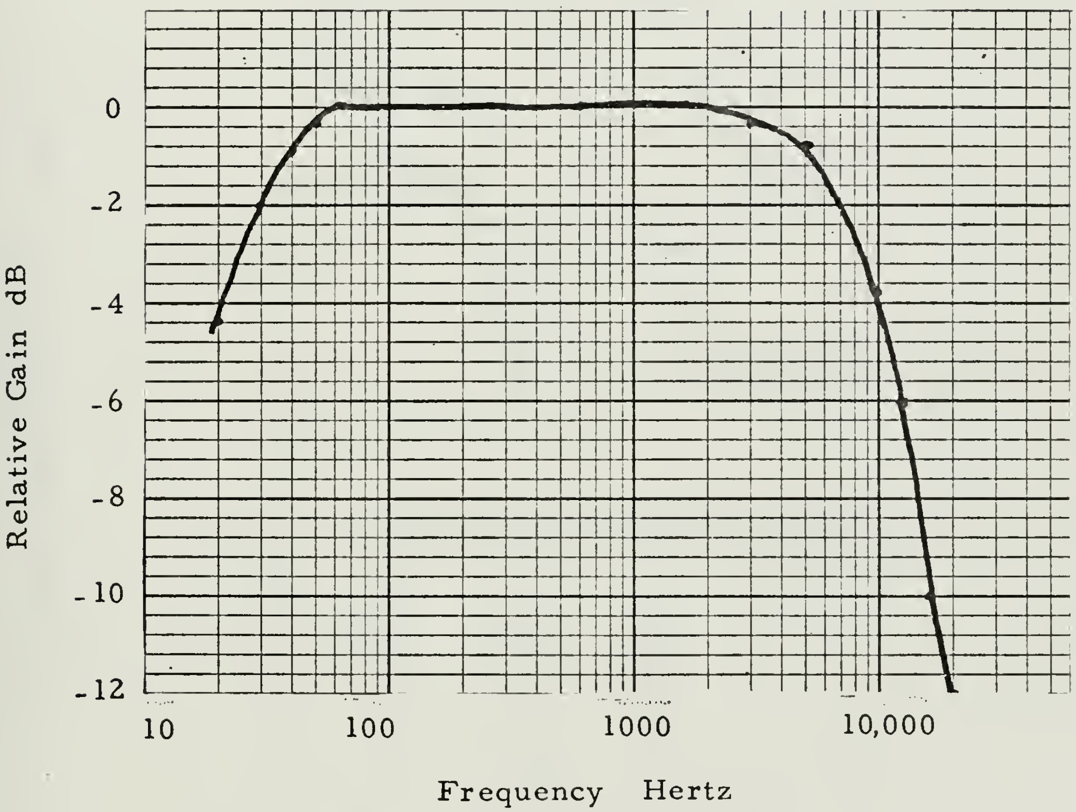
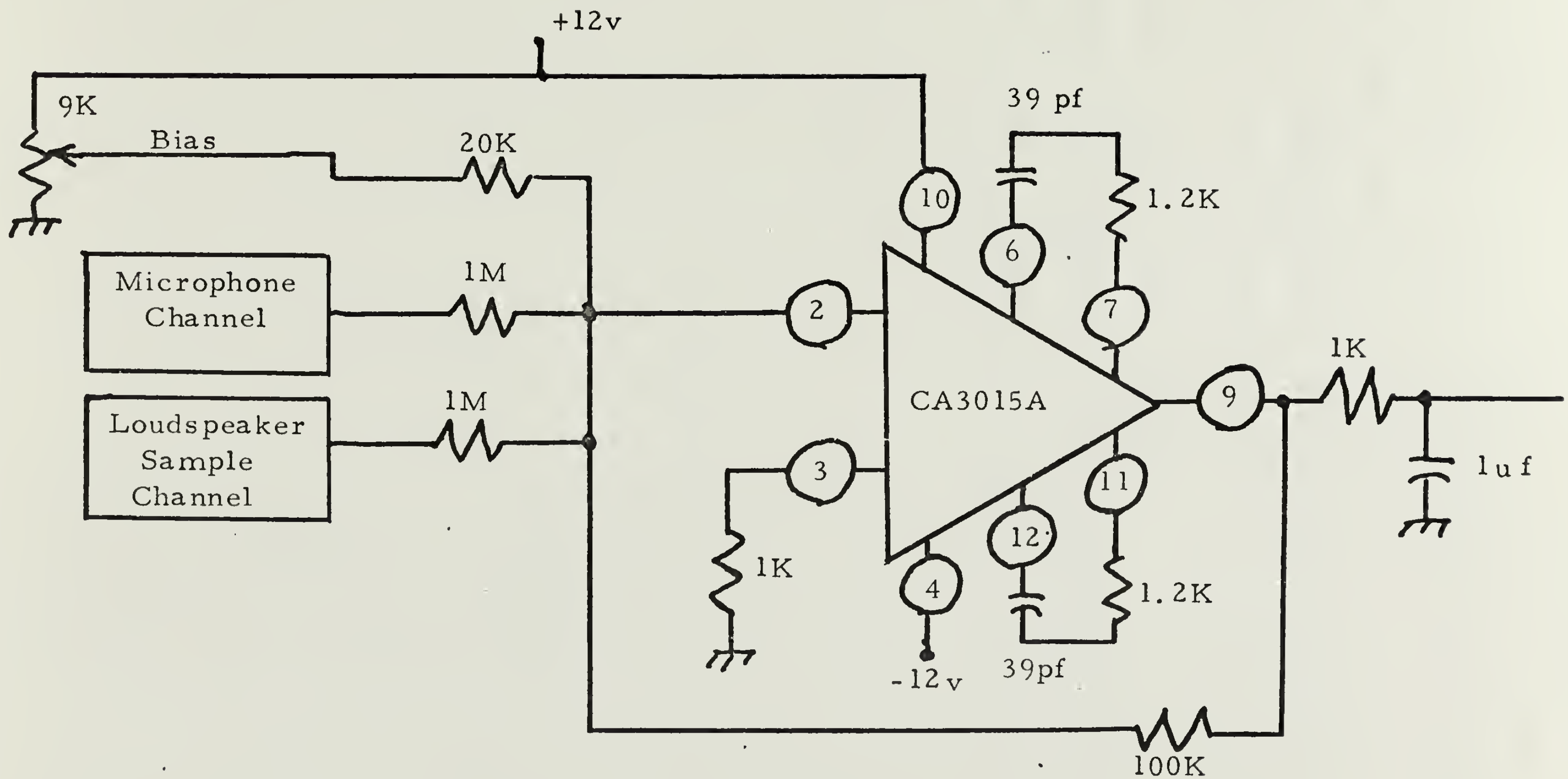


Figure 12
Schematic Diagram of the Scaling Adder



less negative by the presence of noise and driven more negative by the presence of signal. The divide-by-ten scaling is necessary to reduce the large voltage variation to less than one volt, the range over which the control stage AGC operates.

The output is coupled through a low-pass filter consisting of a 1 kilohm resistor and one microfarad capacitor. This provides isolation between stages to reduce feedback.

The additional RC components connected to the operational amplifier provide for phase compensation within the amplifier to ensure stability. [2]

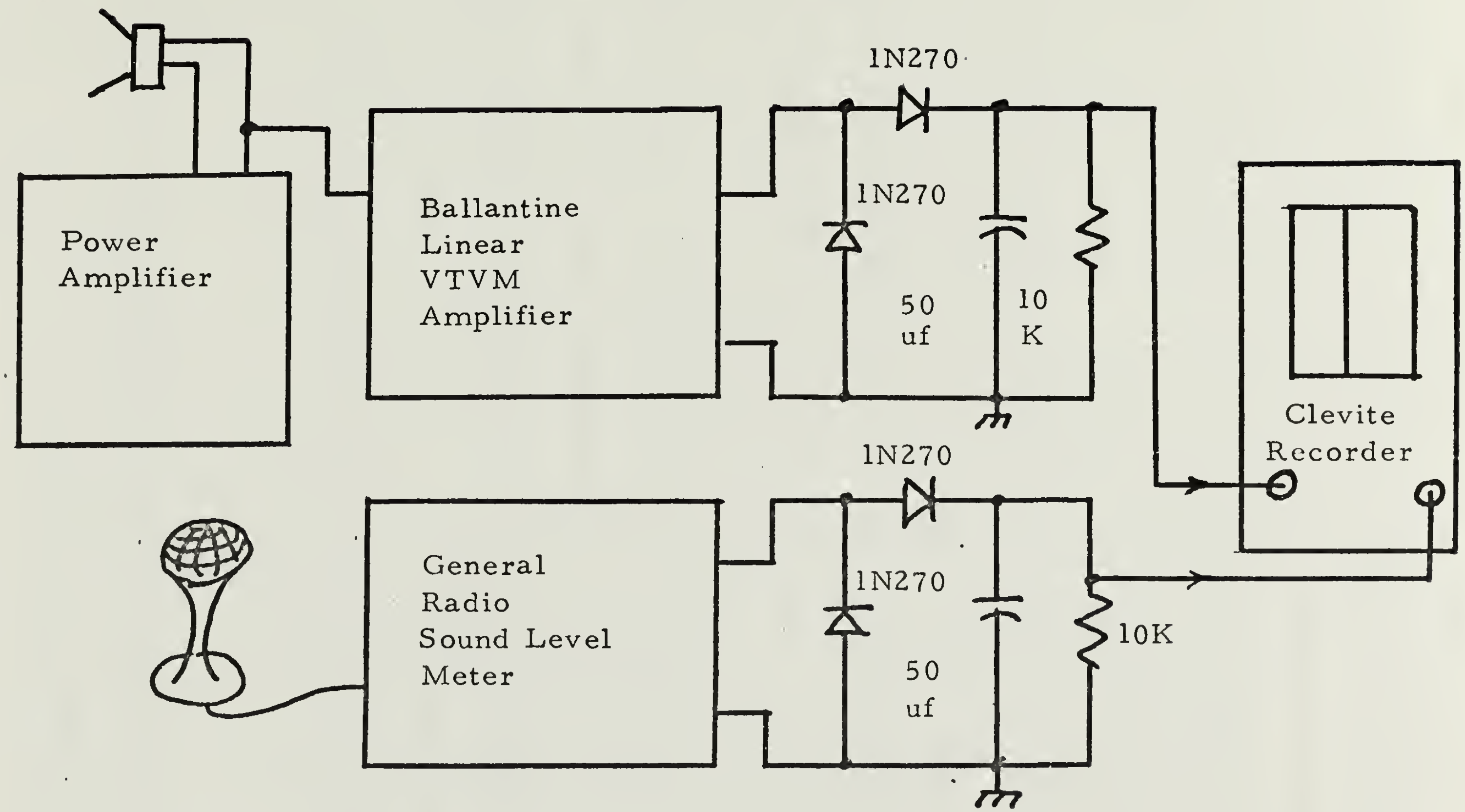
V. SYSTEM OPERATION

A. TEST PROCEDURE AND EQUIPMENT

The design of a simple testing method to describe the subjective results obtained with the use of the automatic control system was difficult, particularly when the signals were obtained from a broadcast radio program. Speech broadcast over a radio has a high peak-to-average energy ratio. The peak-to-average energy ratio for a musical program is not as high. For recording the response of the system, fast time constants were needed to demonstrate the attack time constant of the system. But too fast a time constant resulted in an unreadable recorder trace.

The results in figures 14 through 17 were obtained with the test set up shown in figure 13. The GENERAL RADIO Model 759-B Sound Level Meter was used in the broadband response mode. Its output gave a measure of the broadcast program plus the background noise level. Its input microphone was located one foot in front of and slightly below the control-system microphone. The output from the sound-level meter was an amplified version of what its microphone detected. This output was passed through a peak detector whose release time constant was provided by the 50 microfarad capacitor and 10 kilohm resistor (0.5 second). This time constant provided sufficient filtering to produce a readable recording and yet was fast enough to follow the behavior of the system.

Figure 13
Test Equipment Used in Determining Transient Response
of the Control System



The broadcast level was determined by sampling the signal to the loudspeaker and passing it through a BALLANTINE Linear VTVM Amplifier Model 311, then detecting with a peak detector identical to that used with the noise amplifier. These two signals were then recorded on a CLEVITE BRUSH Recorder Mark 220.

Calibration of the vertical response on the recorder tapes was chosen to be decibels relative to the quiet room level as read by the sound level meter. The characteristics of the room were acoustically hard with many obstructions to provide multiple reflections. With an oscilloscope and the recorder running, the otherwise quiet room level indicated 72 dB on the sound level meter. The other trace markings in dB were determined by broadcasting and recording a music program while reading the average level indicated on the sound level meter. The levels indicated were not exact but were close enough to provide an indication of the subjective results obtained with the use of the control system. The scale on the vertical axis of the recorder tape was approximately linear in dB. The vertical scale on the signal trace is relative to the zero signal level to the loudspeaker.

The noise source was provided by a tape-recording of freeway automobile noise as recorded from the sixth floor of a building about 100 yards from the freeway.

The control system was connected into a HALLCRAFTERS Receiver Model SX130 by disconnecting the tap wire from the volume

control. The sample channel and microphone channel gains were then balanced for the acoustic properties of the room.

B. TEST RESULTS

1. Slow Variation In Noise Level

In order to demonstrate the action of the control system and how it affects the speaker level, test involving changing noise levels were conducted. Figure 14 demonstrates the action of the control system to a slow build up in noise level. The two responses started at the minimum comfortable listening level which recorded at the room noise level of 72 dB. As the noise level was slowly increased by increasing the output level of the noise recording, the signal speaker volume level also increased. This increased the signal-plus-noise level at a more rapid rate. But the speaker level increased nearly linearly as the noise was linearly increased. The faster apparent rate of rise in the signal-plus-noise channel was due to the increased speaker level. Note that the speaker level did not continue to rise after the noise level reached its maximum. This demonstrated that the system was stable.

At the trailing edge of the trace the noise source was rapidly cut off. Both signals dropped with the speaker level falling more slowly than the noise channel level. The spike response in the speaker channel was due to a burst of radar interference.

Figure 14
Response of the System to a
Slow Increase in Background Noise Level

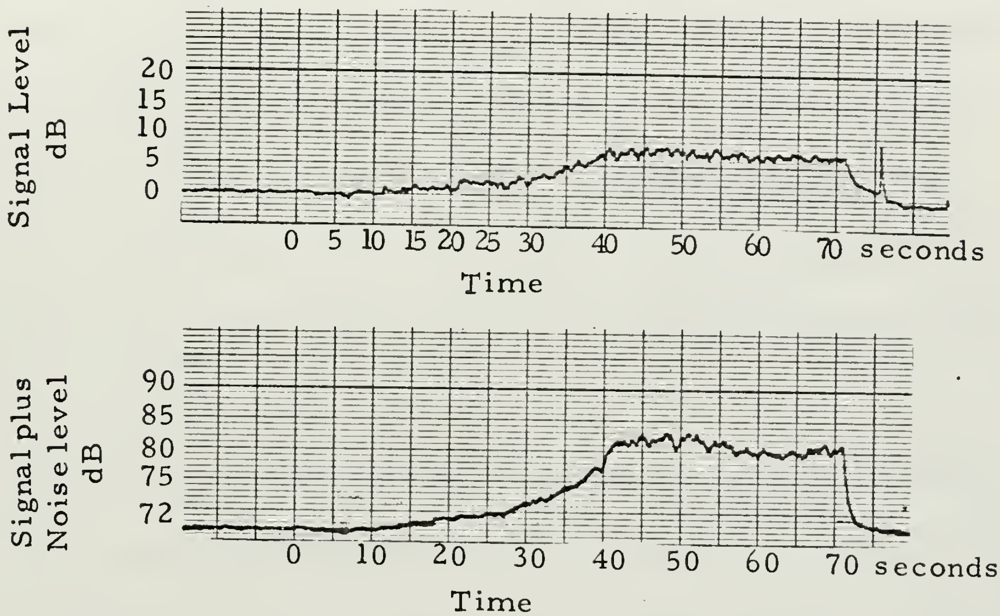
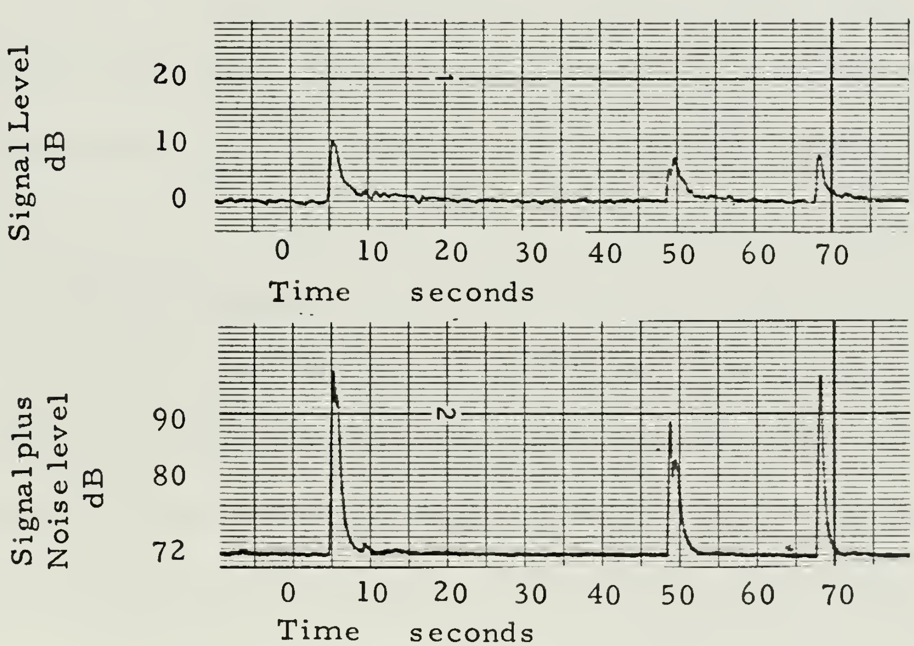


Figure 15
Response of the System to Impulse Noise



2. The Impulse Response

Figure 15 shows the response of the system to impulsive noise. The impulse source was a small empty trash can dropped from 3 feet onto the stone tile floor. The first noise spike recorded well over 90 dB, yet the system was driven up in volume only for a short duration. The transient died out in 3 seconds. This demonstrated that a long-term noise source was necessary to maintain a high speaker level. The output signal level does not fluctuate following transient noise.

For the second impulse the trash can was allowed to bounce twice. The length of the impulse noise was therefore longer, actually peaking at the first bounce. This indicated another property of the system, sensitivity to impulsive noise at a low repitition rate. Slow gun fire rates (30-60 rounds per minute) would keep the speaker level at a higher output level. This would be desirable due to the ear's reduced sensitivity in the presence of loud repetitive impulsive noises.

3. The Step Response

To demonstrate the attack and release time constants a step noise response was recorded in figure 16. The noise recorded jumped to nearly its maximum value in less than 0.5 second while the speaker program did likewise. An additional full second was required for the noise recording to indicate a steady state. The peaked response on the speaker channel was due to the content of the

voice program broadcast. Each peak occurred on an explosive syllable. This example shows clearly the high peak-to-average energy ratio nature of speech. The other programs in the test were voice-music programs. The noise recorded varied as much as 8 dB around an average of 92 dB. This was due to a combination of variations in the recorded noise source and the nature of the speech program.

The release time constant demonstrated several characteristics. The noise source was abruptly cut off. The system started to reduce the signal level immediately, taking two seconds to die out to a low level. It held this lower level for several more seconds while slowly returning to the normal level. The slower rate of return to normal allows the ear to recover its sensitivity to the lower signal level.

4. Other Observations

Subjectively it was observed that the audio signal was at all times detectable in the presence of the noise source. The signal was not lost even during the transitions due to maximum step-noise application. Without the system, whenever the noise level increased ten dB above the room reference level (the speaker level remaining constant) the signal became difficult to understand. With noise 13 dB above the normal ambient level the signal was not discernible.

It was noted that when the system was not properly balanced two different effects occurred depending on whether R· M was greater

Figure 16
Response of the System to
Step Noise Input

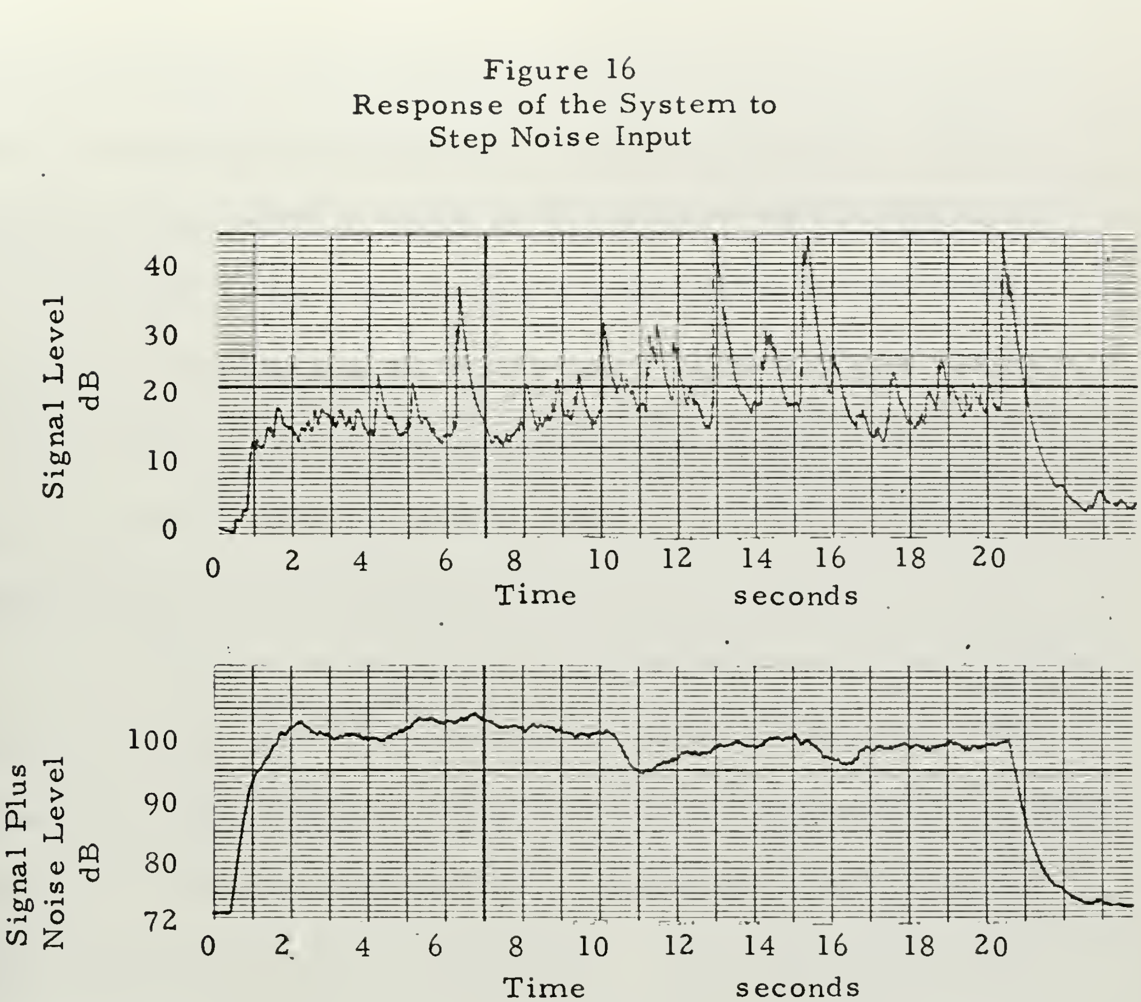
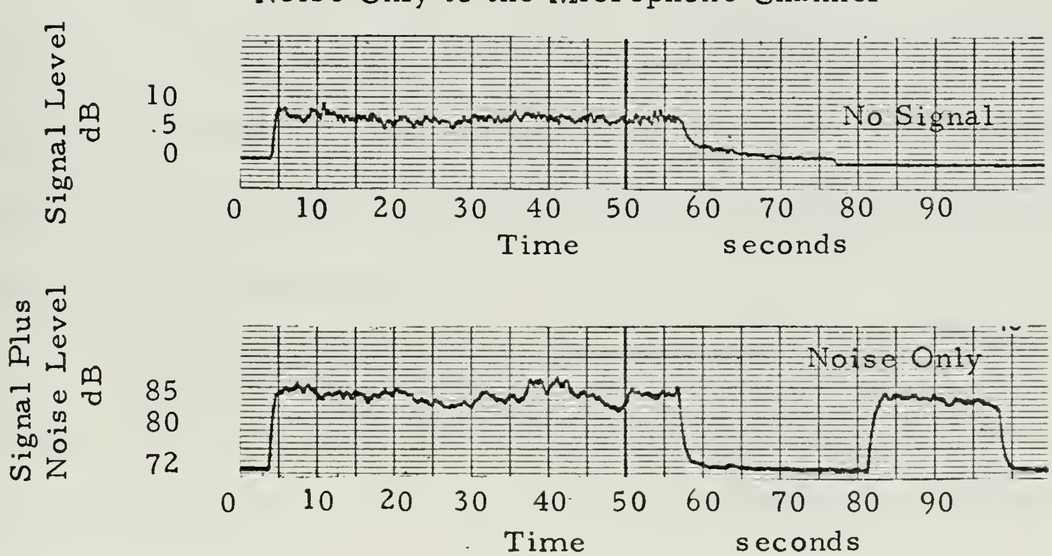


Figure 17
Comparison of Signal plus Noise and
Noise Only to the Microphone Channel



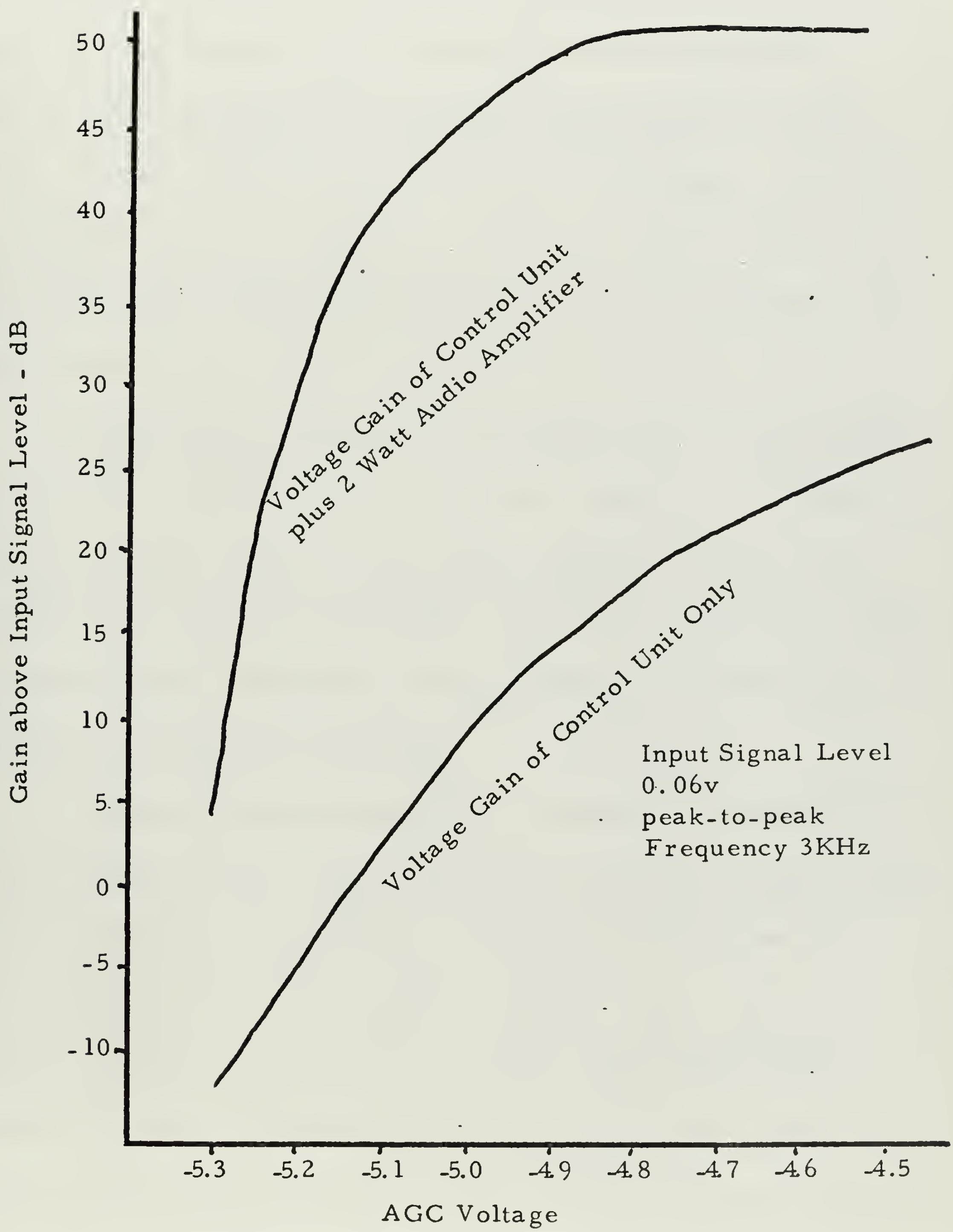
than or less than L. The balance adjustment was not critical, however, and satisfactory subjective response was available when the balance was not perfect. When the system was undercompensated (R· M less than L) the speaker level did not increase sufficiently to maintain a comfortable level. In this case when the noise source was suddenly cut off the speaker would also cut off. When R· M was greater than L the system tended to hunt. When the unbalance was excessive the system immediately operated at maximum volume.

The reader is cautioned when attempting numerical comparison between the indicated levels on the test result figures. The signal-plus-noise channel always contained the loudspeakers influence. Both direct radiation and multipath radiation upon the pickup microphone affected the indicated noise level. The relative gain of the signal channel was calibrated using the sound-level meter.

Figure 17 illustrates this point. The medium noise level produced a satisfactory speaker level. The same noise track was replayed with the speaker turned off. The difference in noise levels indicated with and without speaker signal was less than the random variation of the noise alone.

A final test was made on the total dynamic range of the system. Figure 18 shows the relation between the range of the control unit and the range of the two-watt audio amplifier used in the test.

Figure 18
Dynamic Range of Control Unit and
HALLCRAFTERS Receiver SX130 2 Watt Amplifier



The audio-amplifier curve started above the control-unit curve, the spacing between the curves being equal to the gain of the audio amplifier. The curves are not parallel because the gain of the audio amplifier varied with input-signal level. The gain across the control unit indicated the signal size fed to the audio amplifier. Above an AGC voltage of -4.9 volts the audio amplifier saturated. Larger signals fed into it only increased the distortion. The control unit continued to provide gain up to 31 dB before it saturated. This control unit could adequately control a larger power amplifier, perhaps 25 watts.

The system's range can be extended another 15 dB with the use of another CA3000 in the control stage. However, the 2 watt amplifier produces signals louder than comfortable and a 25 watt amplifier in a closed space becomes unbearable at high power levels. A communications installation in the presence of such loud background noise would not be feasible.

Where several communications channels are monitored by different operators in the same room, the use of the automatic control system could cause each channel to compete for the loudest level. Each channel would sense the other channel as noise, and whenever signal was present on two or more channels each would attempt to override the other. Multiple circuits in the same room can be handled by the control system only under certain circumstances. If each circuit is served by a common control unit and power amplifier

the system would be stable (only one circuit would be permitted to broadcast at a time). An example of this type system is the internal communications system on a ship where one amplifier at one position can receive several different stations, but only one at a time. If several circuits are serviced by their own control unit and power amplifier, they can be used in the same space only if they are not acoustically coupled. This is not a practical situation.

IV. CONCLUSIONS

The automatic control system presented in this paper was capable of controlling the level of a loudspeaker to provide a satisfactory volume level in the presence of a variable background noise level. The response of the system was stable, had a quick reaction to increased noise level, and maintained a satisfactory level decay rate after the noise disappeared. Such a system can be useful in a variety of military and civil applications.

On board a Coast Guard 40 foot and 44 foot patrol boat the coxswain functions as the helmsman, throttleman, and radio operator. The engine compartment is located only a few feet away from his watch-standing position. Variations in engine speed during maneuvering can easily mask his radio receivers. An automatic control unit could be of great value here.

On the bridge of most all ships there are times when conditions require many persons to be present, all trying to communicate with one another and with other locations on the ship. Gunfire and fog horns drown out communications for bridge personnel. A control system would assist in the monitoring of bridge receivers.

In the combat information center again many people can raise the ambient noise level. A control system could be helpful there.

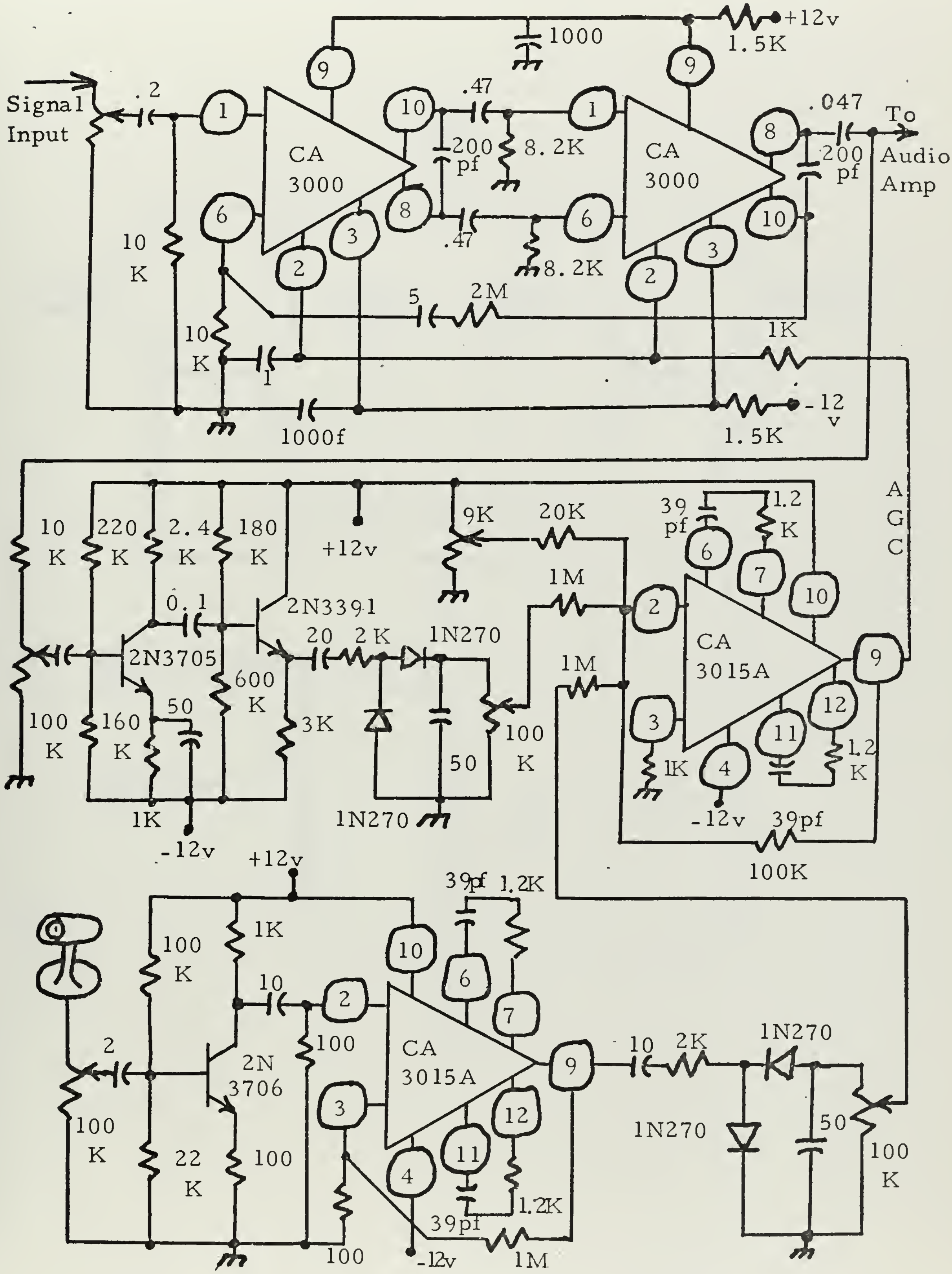
In a lecture hall or classroom the control system would be helpful in a public address system, particularly when jet aircraft

fly overhead. The lecturer would not have to stop and wait for the plane to pass so that he could be heard.

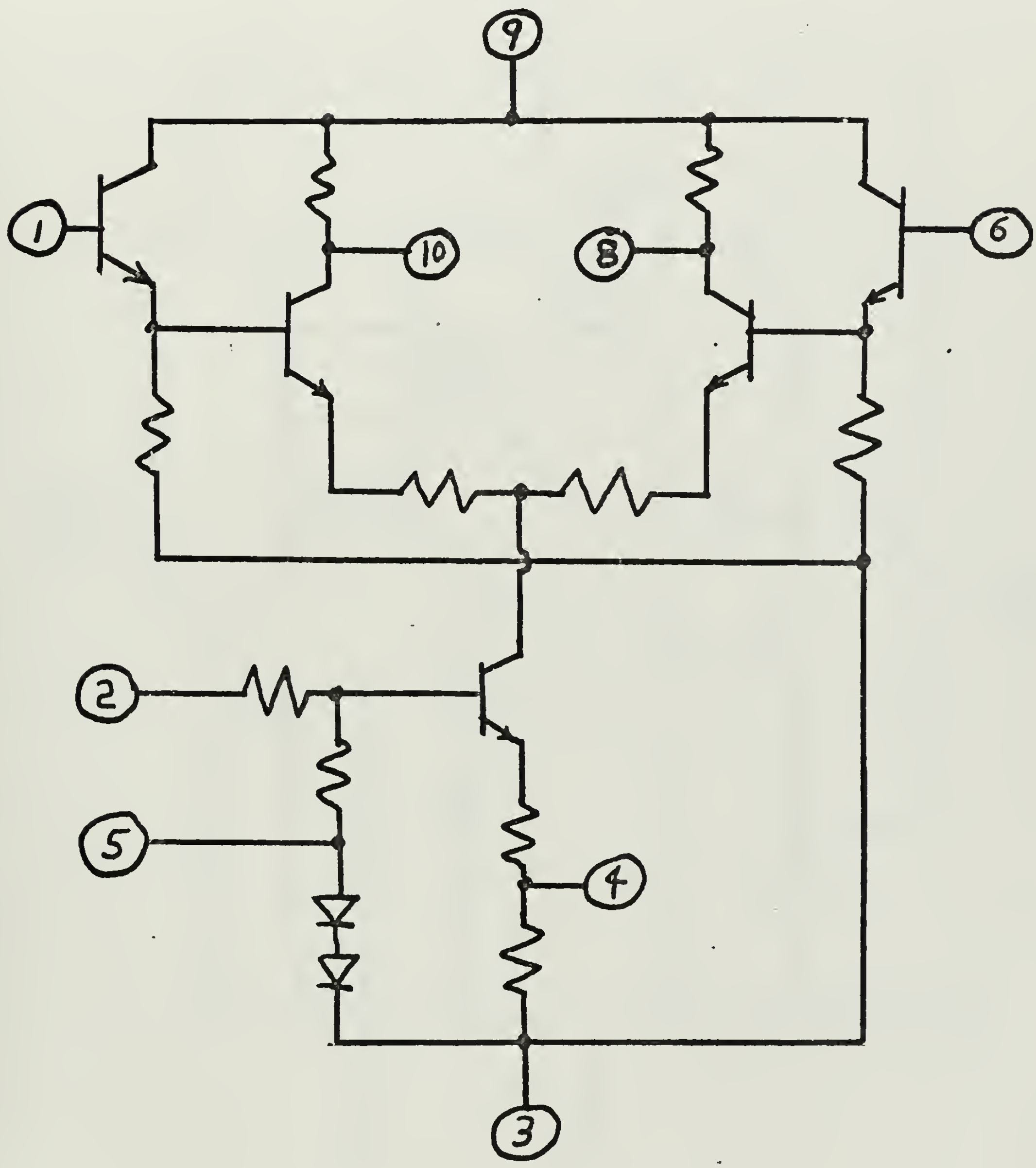
In an automobile the control system would eliminate the adjustment of radio or tape program level due to the change in noise level caused by a passing truck, a blasting horn, or changing automobile speed.

The automatic control system could have application anywhere the background noise level frequently varies and it is desirable to monitor voice communication circuits.

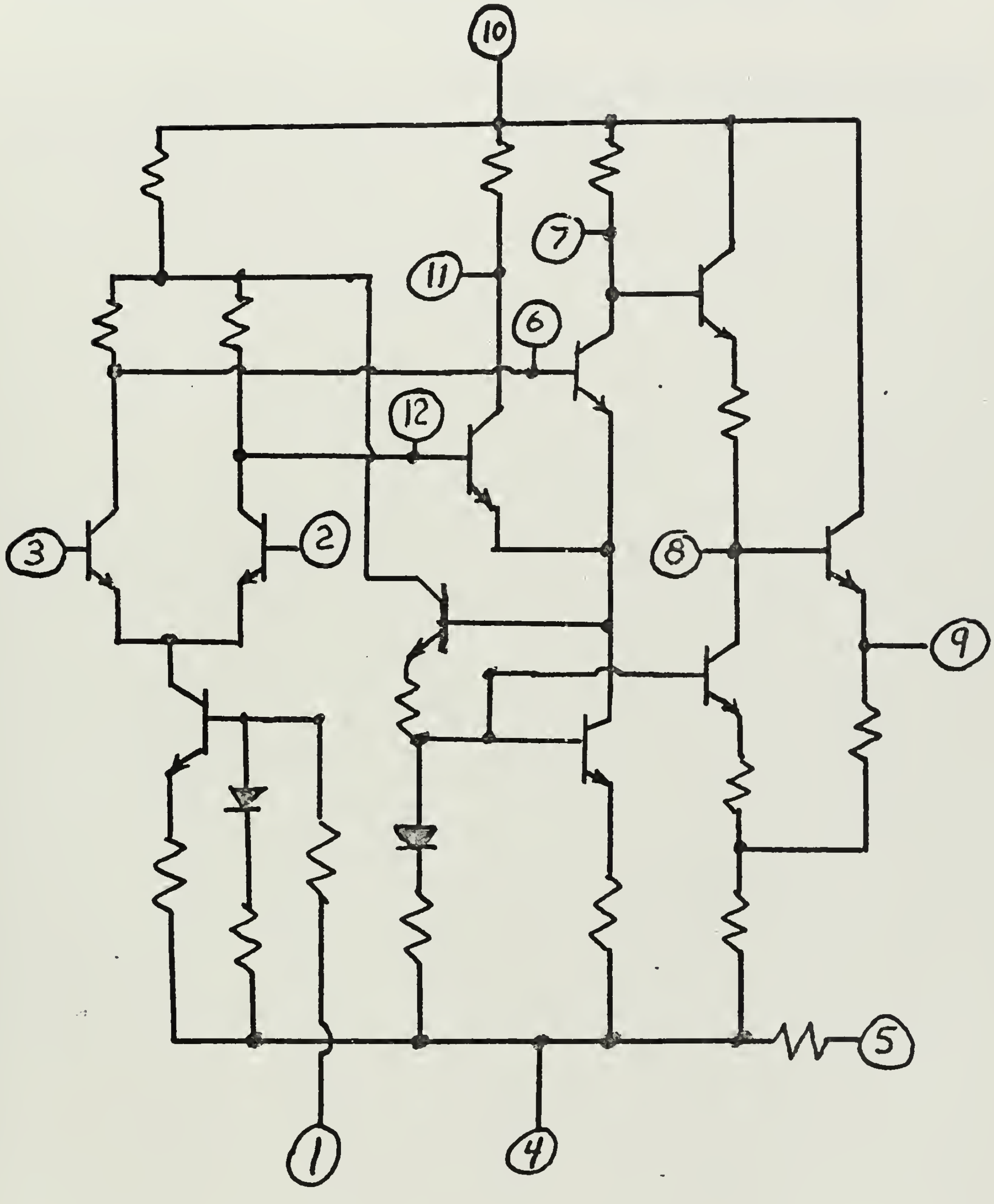
Appendix A. Schematic Diagram of the Automatic Control System. All capacitors in μf except where noted, all resistors in ohms



Appendix B Schematic Diagram of the CA3000 Integrated Circuit



Schematic Diagram of CA3015A
Integrated Circuit



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13. ABSTRACT

Communications operators who must use receivers in a noisy audio environment without earphones are plagued by the effect of a varying background noise level. Often the operators are simultaneously performing other tasks thus can only monitor voice circuits. As the noise level varies the operator must repeatedly adjust the volume control for a comfortable listening level.

An analysis of the physical situation is presented. An automatic feedback control system that measures the predominate range of frequencies of the background noise and automatically adjust the volume control of a power amplifier for a satisfactory listening level is developed. This system relieves the operator of the task of manual volume control and reduces the effect of noise masking.

14. KEY WORDS	LINK A		LINK B		LINK C	
	ROLE	WT	ROLE	WT	ROLE	WT
Automatic Volume Control						
Background Noise						
Audio Noise Interference						
Communications Receiver						

Thesis 125728
H63 Hobson
c.1 Noise activated
automatic volume control.